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Pulse sound: a system of television sound broadcasting using pulses in the video waveform

by

J. R. SANDERS, M.A.(Oxon)
(Research Department, BBC Engineering Division)

BRITISH BROADCASTING CORPORATION

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FOREWORD

This is one of a series of Engineering Monographs published by the British Broadcasting Corporation. About six are produced every year, each dealing with a technical subject within the field of television and sound broadcasting. Each Monograph describes work that has been done by the Engineering Division of the BBC and includes, where appropriate, a survey of earlier work on the same subject. From time to time the series may include selected reprints of articles by BBC authors that have appeared in technical journals. Papers dealing with general engineering developments in broadcasting may also be included occasionally.

This series should be of interest and value to engineers engaged in the fields of broadcasting and of telecommunications generally.

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PULSE SOUND: A SYSTEM OF TELEVISION SOUND BROADCASTING USING PULSES IN THE VIDEO WAVEFORM

SUMMARY

A method of including the sound signal within the video waveform is described which has the advantage that a separate sound transmitter is no longer required; the elimination of the sound transmitter allows the full bandwidth of any radio-frequency channel to be used by the vision transmission.

The performance of such a system is shown to be adequate, both in its ability to withstand adverse reception conditions and in the quality of sound reproduction obtainable.

1. Introduction

1.1 *The Need to Conserve Channel Bandwidth*

In order to broadcast the present British 625-line video signal having a bandwidth of 5·5 MHz together with an f.m. sound transmission, radio-frequency channels of 8 MHz bandwidth have been allocated in the u.h.f. bands IV and V. The extension of the 625-line system to v.h.f. bands I and III would result in a serious loss of coverage unless a saving in channel bandwidth requirements could be effected. In each of the existing v.h.f. 5 MHz channels a 405-line vision transmission having a video bandwidth of 3 MHz and an amplitude-modulated sound transmission are accommodated; the thirteen channels at present available provide 99·5 per cent coverage of the population for two television programmes. It is highly desirable that future 625-line transmissions in these bands should provide a similar coverage, as the cost of 'filling-in' using u.h.f. channels could be high, even if a sufficient number of u.h.f. channel allocations were available.

The conventional sound carrier accompanying the vision signal may be dispensed with if the sound signal is transmitted by means of modulated pulses located within the line-blanking intervals of the video signal; such an arrangement is referred to as a pulse-sound system. A similar system has previously been proposed, but has not satisfied the conditions necessary for broadcasting.[1, 2, 3, 4, 5]

The elimination of the sound carrier provides approximately an extra 1·2 MHz for the vision transmission and it would be possible, for example, to use the present 5 MHz v.h.f. allocations with a 625-line system having a vision main-sideband bandwidth of 4·2 MHz and a vestigial-sideband bandwidth of 0·75 MHz. Such a system would not be generally acceptable since the agreed colour-sub-carrier frequency of 4·43 MHz could not be used. However, the saving of channel bandwidth resulting from the use of a sound-transmission system employing pulses is applicable to television channel allocations of any bandwidth.

1.2 *Required Attributes of the System*

The proposed system must:

(*a*) provide adequate sound quality;

(*b*) withstand poor reception conditions including excessive fluctuation noise accompanying the vision signal, impulsive interference, multipath propagation and co-channel interference;

(*c*) enable an inexpensive decoder to be used in the receiver;

(*d*) cause no significant disturbance to existing receivers.

TABLE 1

Time	*V.H.F. Transmissions*	*U.H.F. Transmissions*	*Receivers*
Present	405:AM	625:FM	Sets on sale: V.H.F. 405:AM U.H.F. 625:FM
Transitional period	405:AM	625:FM + PS	Sets on sale: V.H.F. 405:AM V.H.F. and U.H.F. 625:PS
Final	625:PS	625:PS	All serviceable sets have 625:PS. New sets: V.H.F. and U.H.F. 625:PS facilities ONLY
	← SINGLE STANDARD →		

Key: AM-AM Sound, FM-FM Sound, PS-Pulse Sound

The last requirement (*d*) is termed 'compatibility' and is necessary to facilitate the changeover to a pulse-sound system. Table 1 shows how the transition could be effected.

It will be seen that, during the transitional period, old receivers would receive, as at present, 625 lines with f.m. sound and 405 lines with a.m. sound while receivers on sale would receive 625 lines with pulse sound and 405 lines with a.m. sound. During this period the sound pulses must cause no disturbance to old receivers operating on 625 lines.

At the end of the transitional period all receivers in use would receive only 625 lines with pulse sound; f.m. sound on u.h.f. could then be discontinued.

2. Choice of System Parameters

2.1 *Frequency of Pulses and Restriction of Audio Bandwidth*

As the pulse-sound system is proposed for use with a 625-line system with a line-scan frequency of 15·625 kHz and one pulse is located in each line-blanking interval, sampling of the amplitude of the audio signal takes place at a frequency of 15·625 kHz. Sampling theory states that the maximum reproducible frequency is equal to half the sampling frequency, i.e. 7·8 kHz; in practice this means that an upper frequency limit of about 7 kHz is obtainable from a receiver using a simple filter for the removal of the pulse carrier and the inversion products. This limitation results in a noticeable reduction of the high-frequency content of music when listening under ideal conditions but, in practice, the audio-frequency performance of mass-produced television receivers is such that a 7 kHz upper limit imposes only a small penalty.

2.2 *The Form and Position of the Sound Pulses in the Video Waveform*

This discussion is restricted to position-modulated pulses of fixed magnitude, since this type of pulse train can be shown to provide the best signal-to-noise ratio performance. A suitable value for the deviation of a pulse from its unmodulated position is assumed to be ±1·5 μs and the total duration of the pulse is assumed to be about 1·0 μs; thus, with guard intervals of, say, 0·5 μs at each extreme of the deviation range, an unused interval in the waveform of approximately 5 μs is required.

There are two suitable positions in the line-blanking interval in which the pulse could be placed: (i) the line-synchronizing pulse, and (ii) the 'back porch'. In each case the standard waveform would require some modification.

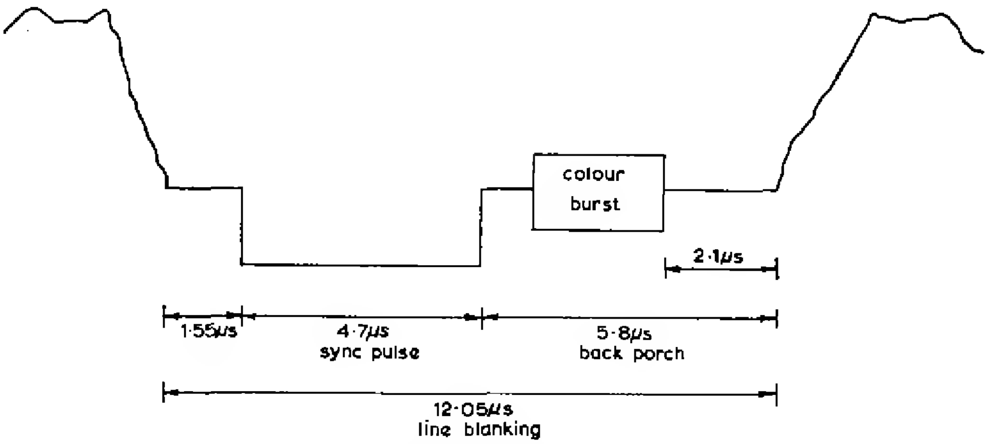

Fig. 1 — Conventional line-blanking interval. All durations nominal

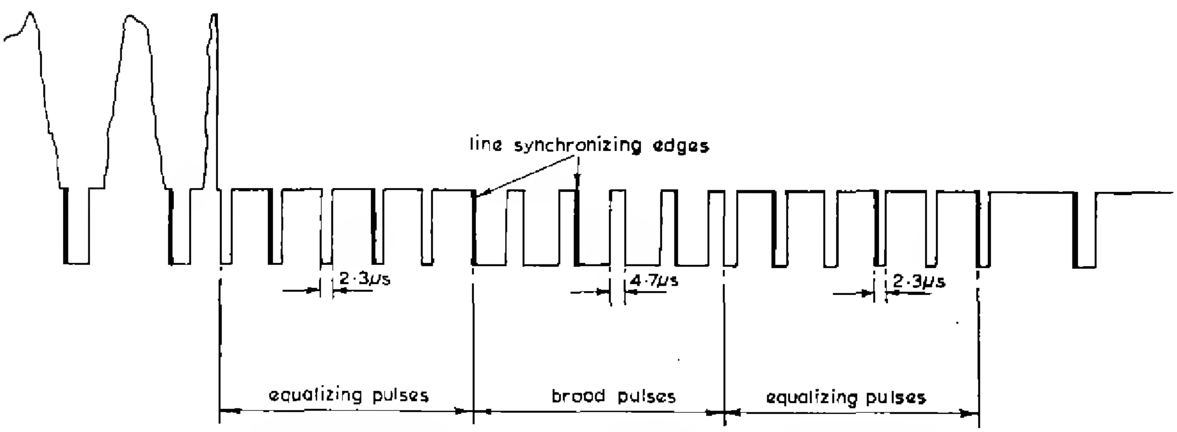

Fig. 2 — Part of conventional field-blanking interval (end of odd fields). All durations nominal

2.2.1 *Sound Pulses Positioned in the Line-synchronizing Periods*

Fig. 1 shows the line-blanking interval of a conventional 625-line waveform. It will be seen that the nominal duration of the line-synchronizing pulse is 4·7 μs, which is quite suitable for a position-modulated sound pulse. However, during the field-blanking interval equalizing pulses of 2·3 μs duration occur in place of line-synchronizing pulses, as illustrated in Fig. 2. Since sound pulses must be regularly spaced and equally deviated this would confine them to the first 2·3 μs of all normal line-synchronizing pulses and would result in inadequate performance due to the restricted deviation. However, this restriction could be overcome by modifying the field-synchronizing waveform to a form in which one field-blanking interval contains no equalizing pulses while, in the other, there is a single equalizing pulse which would not contain a sound pulse;* the modified waveform is shown in Fig. 3.

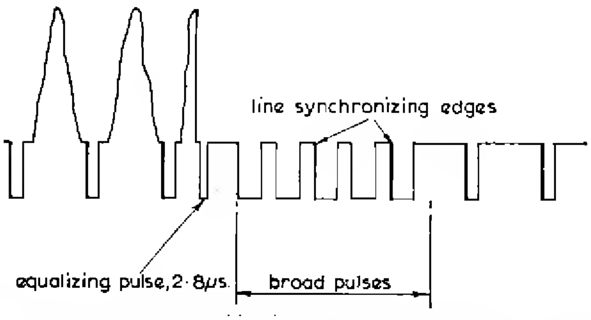

Fig. 3 — Part of modified field-blanking interval (end of odd fields). All durations nominal

The sound pulse could excurse from the bottom of the synchronizing pulse to white level, so that the most effective use of available transmitter power could be made without obtruding beyond the limits of the normal video waveform.

Since it seemed probable that this position of the sound pulses would enable requirements (*a*), (*b*), and (*c*) given in Section 1.2 to be met, it was considered worth while investigating the properties of such an arrangement from the point of view of requirement (*d*), i.e. compatibility with existing receivers.

The pulses used for the investigation were of sine-squared shape, each with a half-amplitude duration of 330 ns, as shown in Fig. 4; the first zero in the spectrum of such a pulse is at 3 MHz. Each pulse was deviated ±1·5 μs from its mean position in the centre of the line-synchronizing pulse and had a magnitude equal to that of the composite video waveform.

Two effects on receivers were observed:

(*a*) Owing to insufficient or non-esixtent line-flyback suppression, sound pulses were visible during the line-flyback trace on some receivers, appearing as a narrow white line down the picture; this line was particularly evident when black level was used as a picture signal and when the pulses were modulated.

(*b*) When deviated by low-frequency audio signals the sound pulses caused position modulation of the raster. The reason for this is that the phase discriminators of line-flywheel circuits respond to the mean position of a number of line-synchronizing pulses and this means position is changed when the pulses are deviated at low audio frequencies. Pulse modulation by frequencies above about 2·0 kHz caused little disturbance since the time constant of the flywheel circuit is then sufficient to ignore the modulation.

All receivers tested with this form of pulse sound showed severe disturbance of synchronization and few receivers were free of the narrow white line caused by the sound-pulse modulating the cathode-ray-tube beam during line-flyback. Since sound pulses in the line-synchronizing period are clearly not compatible with existing receivers the alternative arrangement was investigated.

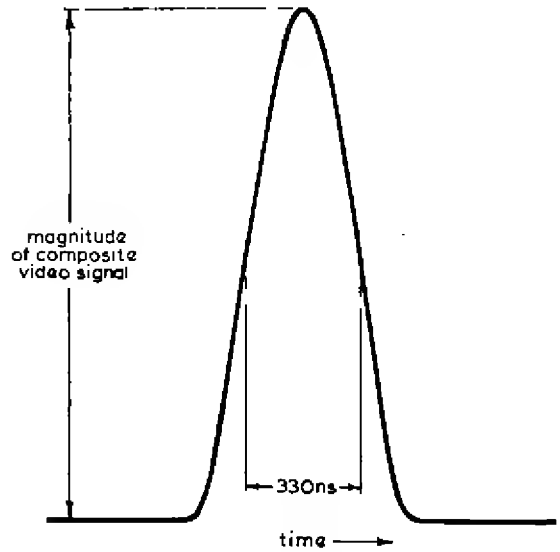

Fig. 4 — Sine-squared pulse

2.2.2 *Sound Pulses Located in the Post-synchronizing Line-blanking Intervals (i.e. the Back Porches)*

Fig. 1 shows that each back porch has a duration of 5·8 μs. However, if the waveform contains an NTSC or PAL colour burst this prevents the use of the first 3·7 μs of the interval, leaving only 2·1 μs available for the sound pulse. It was decided that it would be justified to extend each back porch by 3·5 μs at the expense of picture information. This reduction in active-line period could be accommodated by a change of aspect ratio from 4:3, as used at present, to 5:4. With most present-day receivers, which have 5:4 screens, this change would have no effect upon the quality or appearance of the received picture.

It will be appreciated that, in order to accommodate the sound pulses, the conventional waveform requires modification during the broad-pulse group since a black-level signal corresponding to the back porch does not exist during the broad pulses. Fig. 5 shows the necessary additional pulses added to the broad pulses; these pulses are termed 'platform' pulses. It is necessary to add platform pulses to

* Proposed by G. D. Monteath and A. V. Lord.

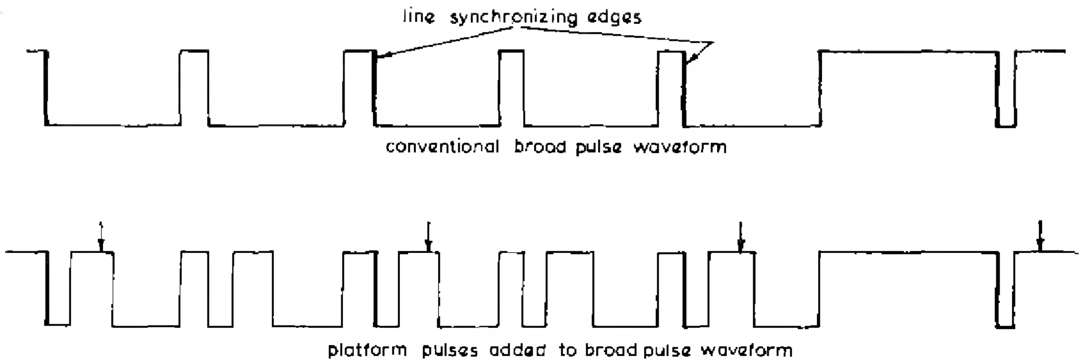

Fig. 5 — Conventional and modified broad-pulse waveforms (end of odd fields) ↓ Position of back-porch sound pulse

all broad pulses in order that odd and even broad-pulse groups remain identical; if this is not the case, receiver interlace is seriously degraded.

In the case of sound pulses located in each back porch, the use of a single equalizing pulse is not strictly necessary. However, it may benefit pulse-sound decoders which use flywheel-type oscillators locked to line-synchronizing pulses to provide time-reference pulses; the design of a flywheel oscillator which is free of phase modulation during the field-sync waveform is thereby made easier.

Fig. 6 shows the chosen form of pulse and its position in the waveform. It is assumed that negative modulation of the vision transmitter is used (i.e. tips of syncs representing peak vision-signal power from the transmitter); this enables a bipolar pulse to be used. For reasons discussed later a pulse with equal excursions about black level is particularly advantageous and such a pulse having an amplitude equal to the excursion from black to white has been adopted. In practice the very low duty factor of the excursion beyond sync level should enable the pulse to be produced by a brief increase in peak transmitter power without any increase in average power. The precise shape of the pulse is shown in Fig. 7, and its spectrum in Fig. 8. As regards shape and spectrum, the pulse may be considered to be the result of subtracting, from one sine-squared pulse shaped as in Fig. 4, a second pulse of the same shape which is delayed, from the first, by approximately 90 ns.

Some other aspects of the choice of parameters for the system will now be discussed.

Sound pulses are least visible on existing receivers if they occur after the receiver line-scan has completed its flyback and before the start of the active line period. In a typical receiver, flyback ends about 5·8 μs after the trailing edge of the line-synchronizing pulse; that is at the time when the conventional active line period would begin. Allowing for a deviation of ±1·5 μs the ideal unmodulated position of the pulse should therefore be 7·3 μs after the trailing edge of the line-synchronizing pulse; in such circumstances the pulse should never be visible providing picture-width and line-shift are correctly adjusted.

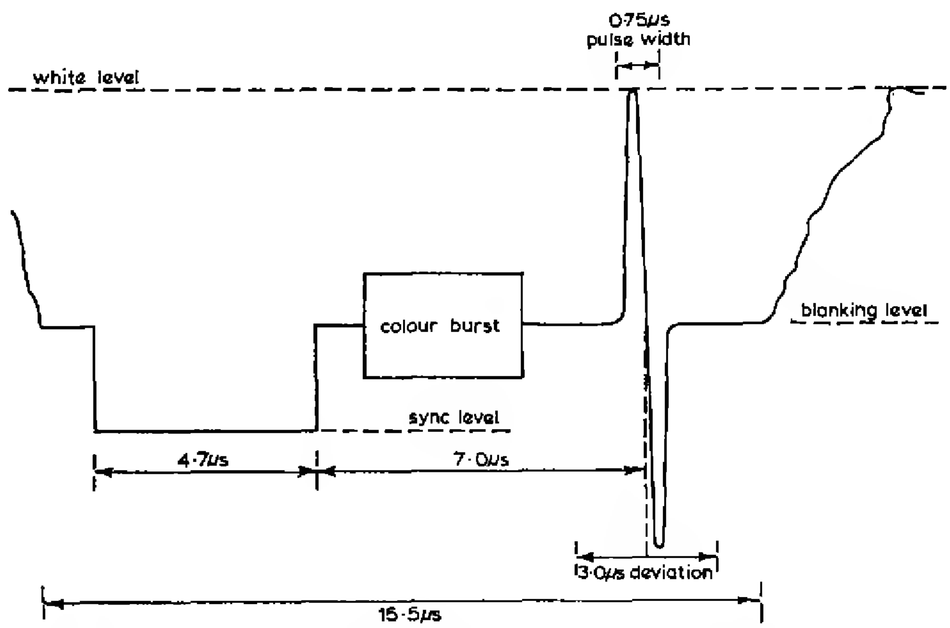

Fig. 6 — Bipolar sine-squared pulse on the back porch. All durations nominal

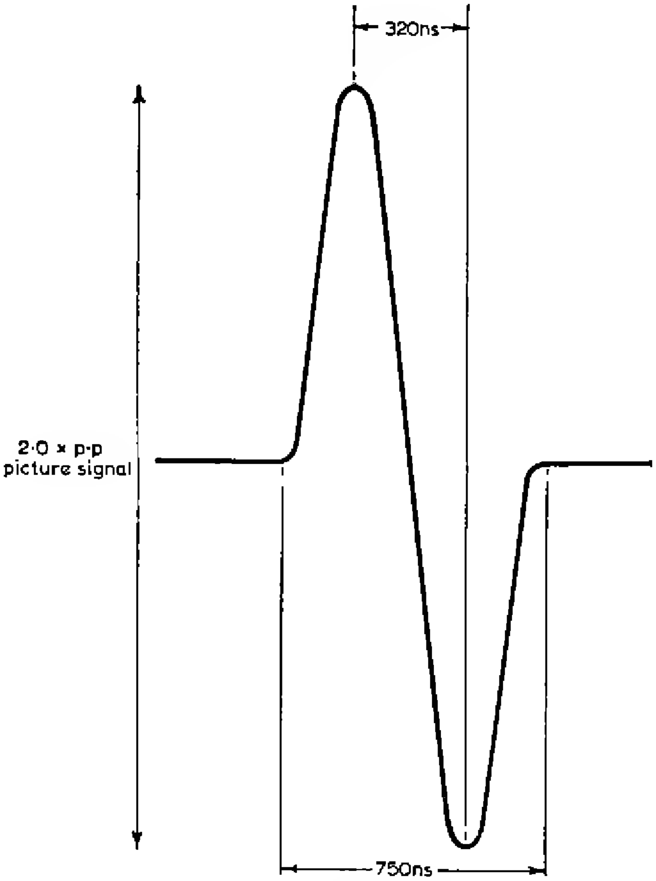

Fig. 7 — Bipolar sine-squared pulse

The position modulation of the receiver raster due to sound pulses disturbing synchronization can be minimized by using pulses of suitable shape and by positioning them carefully with respect to the line-sync pulses. Sync-separator circuits in receivers have relatively little response at frequencies higher than a few hundred kilocycles, so that a sound pulse characterized by a spectrum with maximum energy concentrated around a frequency higher than this will be considerably attenuated. From Fig. 8 it will be seen that, due to the symmetrical shape of the sound pulse, the spectrum has zero energy at zero frequency and maximum energy at about 1·2 MHz. The result of this is that only a small sound pulse, if any, is present in the sync-separator output. The position of the pulse also influences the disturbance of synchronization. Fig. 9 shows diagrammatically the relative timing of a receiver line-scan waveform and the sync-separator output as applied to the phase discriminator of a flywheel deflection circuit; a small residual sound pulse may accompany each sync pulse. Any spurious sampling of the line-scan waveform by a modulated sound pulse will cause changes in the control voltage fed to the timebase frequency-control stage which will result in position modulation of the raster. The smallest variation in control voltage is obtained when the sound pulse is so timed that the spurious sampling occurs at that part of the line-scan waveform having the lowest slope; from this point of view a sound-pulse position about 7·0 μs after the trailing edge of the line-sync pulse is advantageous. The choice of pulse parameters was also influenced by considerations of signal-to-noise performance. This is discussed in the next section.

Tests were carried out on a number of different types of receiver using the form of sound pulse shown in Fig. 7. Of the thirteen different types of dual-standard receiver used none exhibited troublesome positional disturbances of the picture but two were slightly affected when the sound pulses were fully modulated at low frequencies. With four types of receiver maximum modulation caused the deviated sound pulses to be just visible with black level as picture signal, although the scan amplitude and shift controls had been adjusted so that the picture just filled the tube mask.

The small degree of visibility and positional disturbances shown by these tests indicate that the presence of sound pulses would not disturb the large majority of viewers using dual-standard receivers.

The modification to the field block has already been mentioned and takes the form of black-level pulses starting at the times at which trailing edges of line syncs would occur and lasting 9·3 μs. The effect of the platform pulses on receiver interlace and on hold-in range has been studied by means of radiated tests. These show that while 58 per cent of sets exhibit a somewhat reduced range of field-hold there is no significant change in the reliability of field-synchronization and that the quality of interlace is similar to that obtained using the standard waveform.

The platform pulses are of longer duration than are strictly necessary, but confer two possible advantages. First, the design of pulse-sound decoders may be simplified by the presence of an uninterrupted sequence of complete line-synchronizing pulses of constant duration and, secondly, an NTSC or PAL colour-burst sequence could be continued throughout the field-synchronizing period, possibly easing the design of burst-locked oscillators in colour receivers. Should these possibilities not be exploited the duration of each platform pulse could be reduced so as to embrace only the deviation range of the sound pulse.

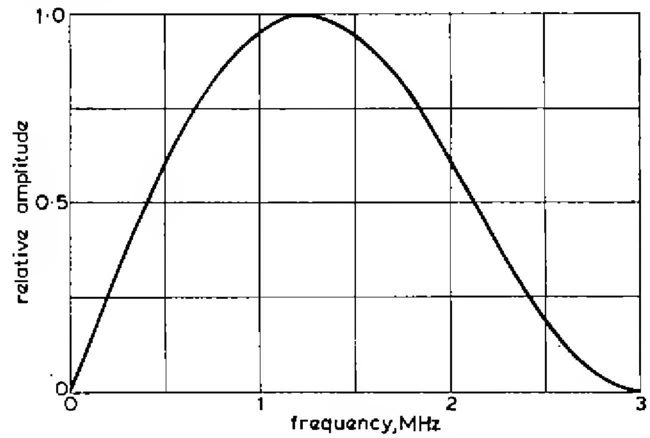

Fig. 8 — Bipolar sine-squared pulse spectrum

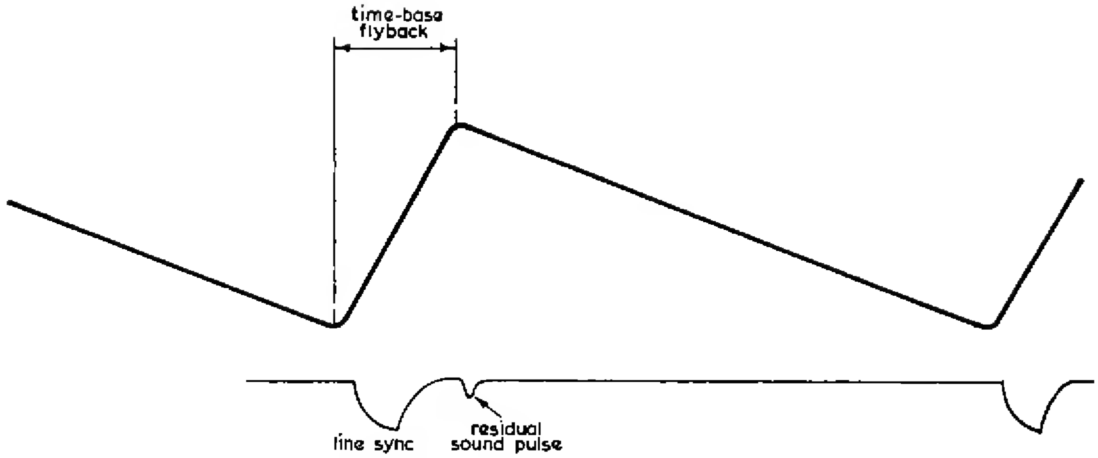

Fig. 9 — Relative timing of sync-separator output and line-scan waveform of a typical receiver

2.3 *Signal-to-noise Ratio Performance*

The signal-to-noise ratio performance of a system using the bipolar pulse described in Section 2.2.2 may be calculated as follows:

If S = the slope in $V/\mu s$ of the pulse edge at the input to the decoder

V_N = the r.m.s. video noise voltage in volts accompanying the pulse

d = the maximum deviation in μs of the pulse from its unmodulated position

ε = the r.m.s. uncertainty in μs in the position of the pulse edge

then:

$$\varepsilon = \frac{V_N}{S}$$

The r.m.s. noise voltage V_N' at the decoder output is therefore:

$$V_N' = \frac{\lambda V_N}{S}$$

where λ is a constant depending on the decoder.

Similarly, the peak audio signal voltage V_s at the decoder output is:

$$V_S = \lambda d$$

Thus the r.m.s. audio signal-to-noise ratio $R_{P.P.M.}$ is

$$R_{P.P.M.} = \frac{\lambda d}{\sqrt{2}} \cdot \frac{S}{\lambda V_N}$$

$$= \frac{dS}{\sqrt{2} V_N}$$

For a composite signal of the form shown in Fig. 6. in which the picture-signal excursion is 0·7 V, the picture signal-to-noise ratio is

$$R_V = \frac{0{\cdot}7}{V_N}$$

and the maximum theoretical slope of the pulse edge (giving the maximum audio signal-to-noise ratio) is

$$S_{max} = 7{\cdot}0 \text{ V}/\mu s$$

This yields a ratio of audio signal-to-noise ratio to picture signal-to-noise ratio of

$$\frac{R_{P.P.M.}}{R_V} = 10{\cdot}6$$

Assuming that the video bandwidth is 5·5 MHz (as in Standard I), the shape of the sound-pulse spectrum, as shown in Fig. 8, permits the bandwidth of the input circuit to the sound decoder to be restricted to 3·0 MHz with a consequent reduction in noise. Under these circumstances:

$$\frac{R_{P.P.M.}}{R_V} = 14{\cdot}3$$

$$= +\ 23{\cdot}1 \text{ dB}$$

Theoretically, an improvement of 3 dB in the performance of the pulse-sound system can be obtained if both lobes of the pulse are used; decoding techniques for obtaining this improvement are discussed later in this report. With a conventional a.m. television sound system having the same audio bandwidth the ratio of the audio to video signal-to-noise ratios, $R_{A.M.}/R_V$, is + 27 dB.

The noise performance of the pulse-sound system deteriorates rapidly below a 'threshold' value of video signal-to-noise ratio; the threshold occurs when the peak values of noise components are comparable with half the peak-to-peak magnitude of the sound pulse. The significance of this effect may be evaluated for the case of a simple decoder which uses only one half of the bipolar pulse. It is necessary for the designer to select a voltage level such that signals exceeding this level will be recognized as a sound pulse; this is referred to as the triggering level. For a composite signal in which the picture-signal excursion is 0·7 V, the triggering level selected would probably be about 0·4 V below black level so that a sufficiently steep part of the pulse is used by the decoder (thus providing nearly the maximum audio signal-to-noise ratio) and yet noise pulses with amplitudes of less than 0·4 V are excluded. The video signal-to-noise ratio at which a significant number of the

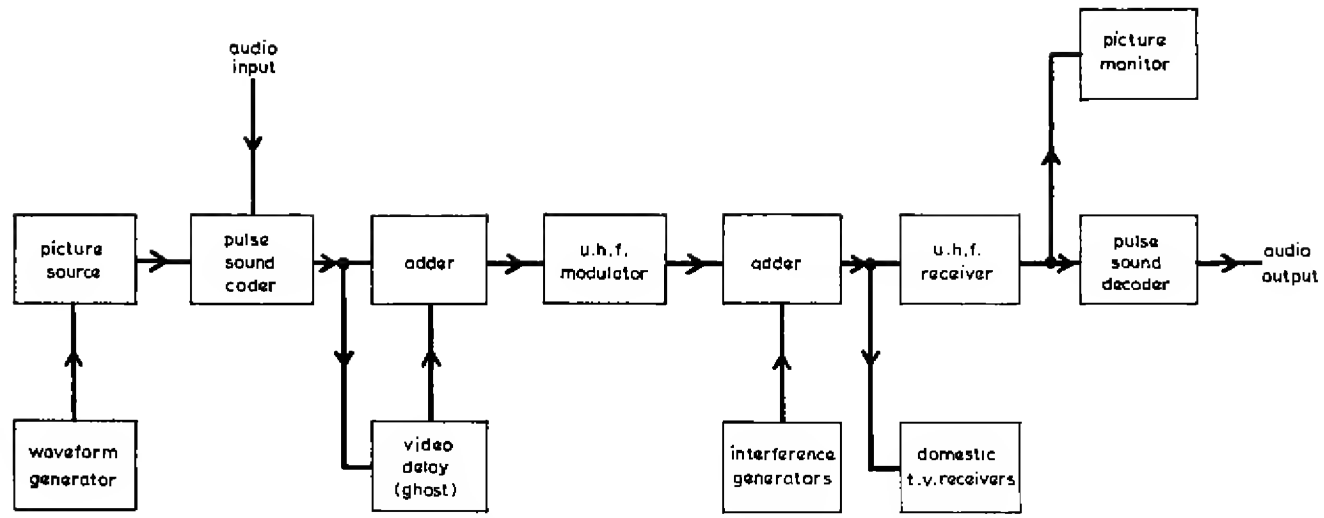

Fig. 10 — Complete pulse-sound experimental arrangement

noise pulses exceed the triggering level is the threshold at which catastrophic failure of the sound begins.

A well recognized assumption is that:

$$\frac{\text{Quasi-peak noise level}}{\text{r.m.s. noise level}} = 12 \text{ dB}$$

At a video signal-to-noise ratio of 14·3 dB* (measured in a 5·5 MHz bandwidth) peaks of noise will be $\pm$ 0·4 V after low-pass filtering to 3·0 MHz. Thus, in this simple decoder catastrophic failure will occur at video signal-to-noise ratios less than 14·3 dB. By the use of both lobes of the pulse the threshold can be lowered to about 13 dB together with a small improvement in the 'above-threshold' performance. Such signal-to-noise ratios for the video signal are well below those normally obtainable anywhere within the service area of a transmitter.

As mentioned in the preceding Section, the choice of pulse shape and spectrum was influenced by signal-to-noise considerations. If the pulse spectrum were widened (i.e. a narrow pulse) the signal-to-noise performance above the threshold would improve, but as the amplitude of noise peaks increases with bandwidth, the threshold would occur at a lower noise level. If the spectrum of the pulse were narrowed, the pulse width would increase and this would degrade the performance above the threshold. A further degradation would occur due to the need to reduce deviation so as to confine the pulse excursions within the allotted time limits. The present figure of 3 MHz is considered optimum. A detailed specification of the composite vision and sound signal is given in the Appendix.

* $14 \cdot 3 \text{ dB} = 12 \text{ dB} + 20 \log_{10}\left(\frac{0 \cdot 7}{0 \cdot 4}\sqrt{\frac{3 \cdot 0}{5 \cdot 5}}\right)$

3. An Experimental Assessment of the System

3.1 *Instrumentation of Coding and Decoding*

A block diagram of equipment used to test the system is shown in Fig. 10.

The pulse-sound coder provided a composite signal which negatively modulated a u.h.f. carrier in accordance with the vision signal characteristics of Standard I, except

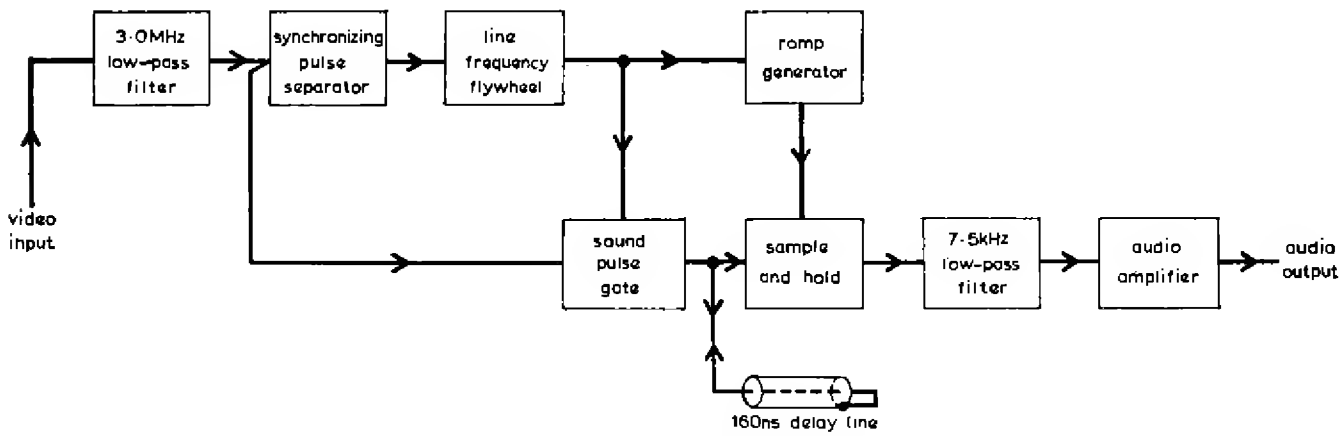

Fig. 11 — Pulse-sound decoder

in so far as pulse sound was added. A u.h.f. receiver demodulated this carrier and provided a video signal for the pulse-sound decoder. In order to test the system under adverse conditions, provision was made for adding a delayed signal or 'ghost' before modulation so as to simulate multipath propagation; in addition flat-spectrum noise, impulsive interference and c.w. interference could be added to the output of the modulator.

The decoder is shown in block-diagram form in Fig. 11. The line-frequency flywheel, having a long time-constant, generated from the incoming line-synchronizing pulses, a train of reference pulses which were unaffected by the presence of noise accompanying the video-signal input.

4 μs pulses from the flywheel were used to gate the sound pulses and extract them from the video waveform. Decoding of the sound pulses was carried out by means of a train of linear ramp waves, derived from the flywheel reference pulses which were sampled by the separated sound pulses in a sample-and-hold circuit. The output of the sample-and-hold circuit was fed to the 7·5 kHz low-pass filter whose output consisted of the recovered audio signal.

In order to provide the best signal-to-noise performance the gated sound pulses were applied to a 160 ns delay line driven from its characteristic impedance and short-circuited at the far end. This delay line combined the positive and negative half cycles of each sound pulse to produce the pulse shown in Fig. 12. The reduced sensitivity to noise conferred by the use of such a delay line was found experimentally to be about 2 dB. From the delay line, the sound pulses passed to a level detector whose output changed abruptly as the leading edge of each pulse reached a particular voltage level, thus defining the time of occurrence of the pulse. This particular level was referred to as the 'triggering level' and was adjusted by means of a pre-set control in order that an optimum setting could be used in the presence of noise. The approximate triggering level at which optimum performance was achieved is indicated in Fig. 12. The level detector generated the pulse used to sample the ramp-waveform described previously. The particular decoder used was designed solely to allow the system performance to be assessed with a minimum of instrumental limitations and would, in consequence, be considered too elaborate for general use in domestic receivers; however, comparatively simple decoders can be constructed with similar performances.

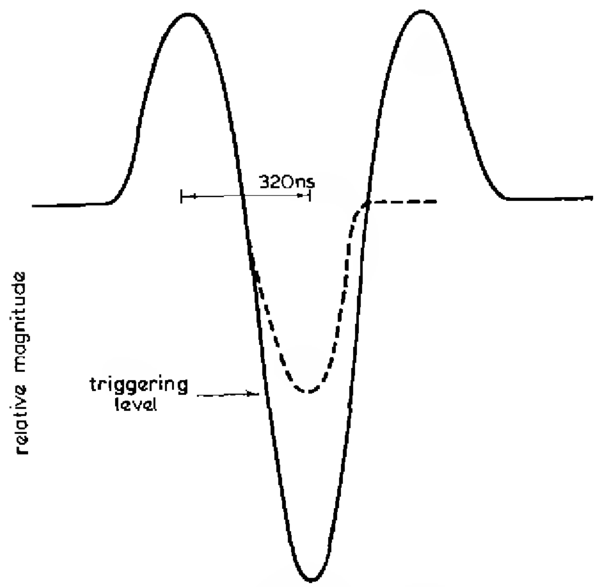

Fig. 12 — The effect of a short-circuit delay line upon the sound pulse

- - - - - Input ——— Output

3.2 *Measured Performance*

The following aspects of performance were measured:

(*a*) Audio-distortion/frequency and audio-response/frequency characteristics under ideal conditions.

(*b*) Audio signal-to-noise ratio in the presence of random noise.

(*c*) Spurious sound signals in the presence of steady c.w. interference.

(*d*) Susceptibility to the presence of an echo.

(*e*) Susceptibility to impulsive interference.

3.2.1 *Audio-Distortion/Frequency and Audio-Response/Frequency Characteristics under Ideal Conditions*

As the coder used produced P.P.M. pulses based upon natural sampling and as the decoder was of the ramp form some harmonic distortion and inversion tones were produced. Fig. 13 shows the calculated and measured values of wanted and unwanted components as a function of modulation frequency when maximum deviation ($\pm$ 1·5 μs) was used. The calculated values have been abstracted from an unpublished report* in which the production of harmonic and inversion tones is dealt with in detail.

It will be seen that there is good agreement between the calculated and measured values. The maximum value of unwanted component occurred with a modulation frequency of 3·9 kHz, when the unwanted component had a frequency of 7·8 kHz and an amplitude equal to about 5 per cent of the wanted 3·9 kHz signal; when smaller values of deviation were used the relative levels of distortion were reduced.

The pulse-sound system had an inherently flat response/frequency characteristic from 0 to 7 kHz. The frequency response below 7·8 kHz was limited only by the characteristics of the low-pass filters necessary in both coder and decoder to remove the inverted modulation spectrum present as the lower sideband of the sampling frequency. (The upper sideband of the sampling frequency and sidebands of higher harmonics of the sampling frequency are also undesirable but need less attenuation as they are inaudible.)

With the arrangement described, the inversion components and harmonic distortion did not produce any significant degradation of the quality of reproduced speech and music.

* By A. V. Lord.

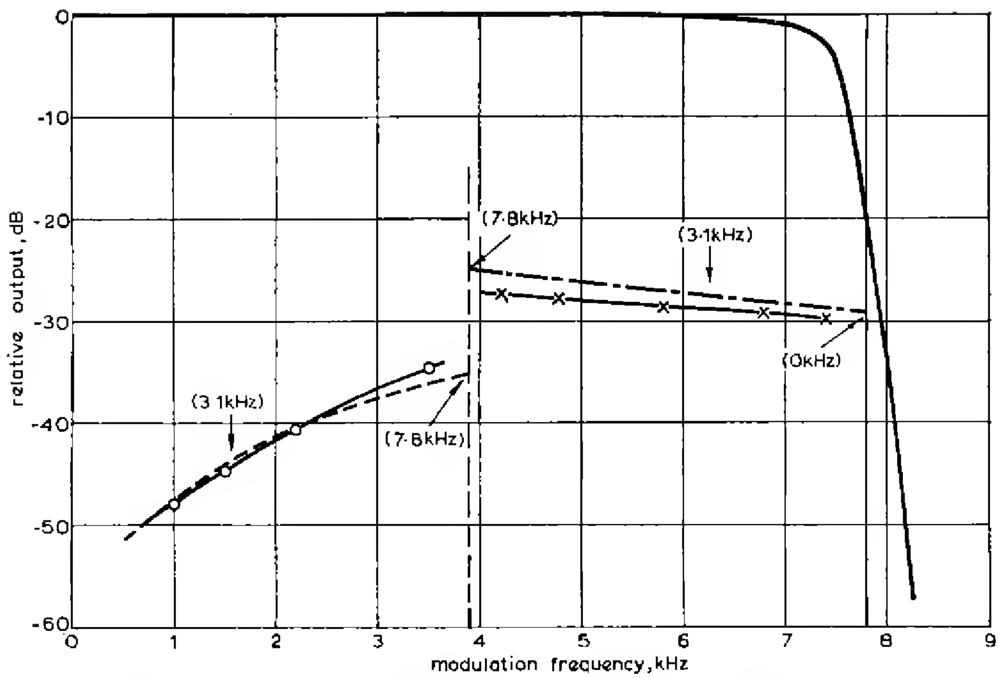

Fig. 13 — Calculated and measured output components of ramp demodulation of pulse position modulation using natural sampling, as functions of modulation frequency

——— Output filter characteristic
—o——o— Measured unwanted second harmonic component
—x——x— Measured unwanted inversion component
— — — — Calculated unwanted second harmonic component
– – – – – Calculated unwanted inversion component
( ) Frequency of unwanted component

3.2.2 *R.M.S. Audio Signal-to-r.m.s. Noise Ratio in the Presence of Random Noise*

Fig. 14 shows how the measured r.m.s. audio signal-to-r.m.s. noise ratio varied with the peak picture signal-to-r.m.s. noise ratio. The peak picture signal-to-r.m.s. noise ratio was measured in a 5·5 MHz bandwidth.

The r.m.s. audio signal-to-r.m.s. noise ratios correspond to 40 per cent of peak pulse deviation and were measured by a full-wave rectifying, amplifier-detector reading mean level and calibrated to indicate the r.m.s. value of a sine-wave input; this type of instrument underestimates the value of white noise by 1 dB in comparison with its true r.m.s. value and this fact has been allowed for in the results. It will be seen that the r.m.s. audio signal-to-r.m.s. noise ratio and peak picture signal-to-r.m.s. noise ratio are linearly related, as would be expected, for values of picture signal-to-r.m.s. noise ratios greater than about 13·5 dB; below this value catastrophic failure occurred.

The measurements show that the difference between audio r.m.s. signal-to-r.m.s. noise ratio and peak picture signal-to-r.m.s. noise ratio is 15·5 dB. In Section 2.3 of this report this difference was predicted to be 26·1 — 8·0 = 18·1 dB if use were made of both lobes of the pulse. Thus the measured performance of the system was 2·6 dB worse than that predicted.

Non-linearity of the curve shown in Fig. 14 at peak picture signal-to-r.m.s. noise ratios in excess of 26 dB is due to inherent noise in the coding and decoding apparatus caused by unwanted phase modulation of the pulses due to instrumental deficiencies and is not fundamental to the system.

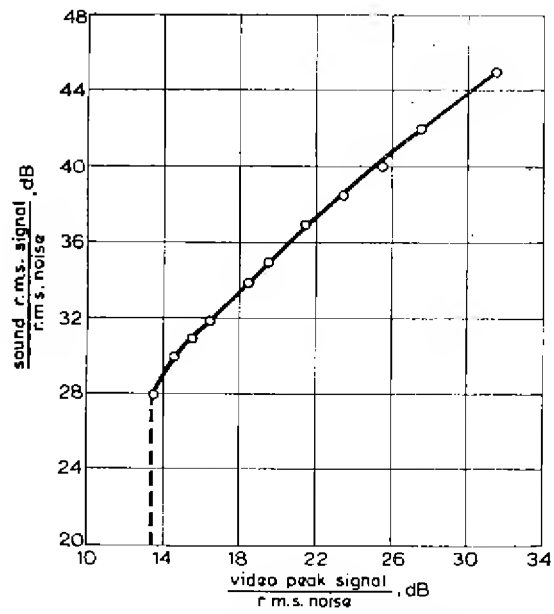

Fig. 14. — The measured relationship between sound and video signal-to-noise ratio. (40 *per cent deviation*)

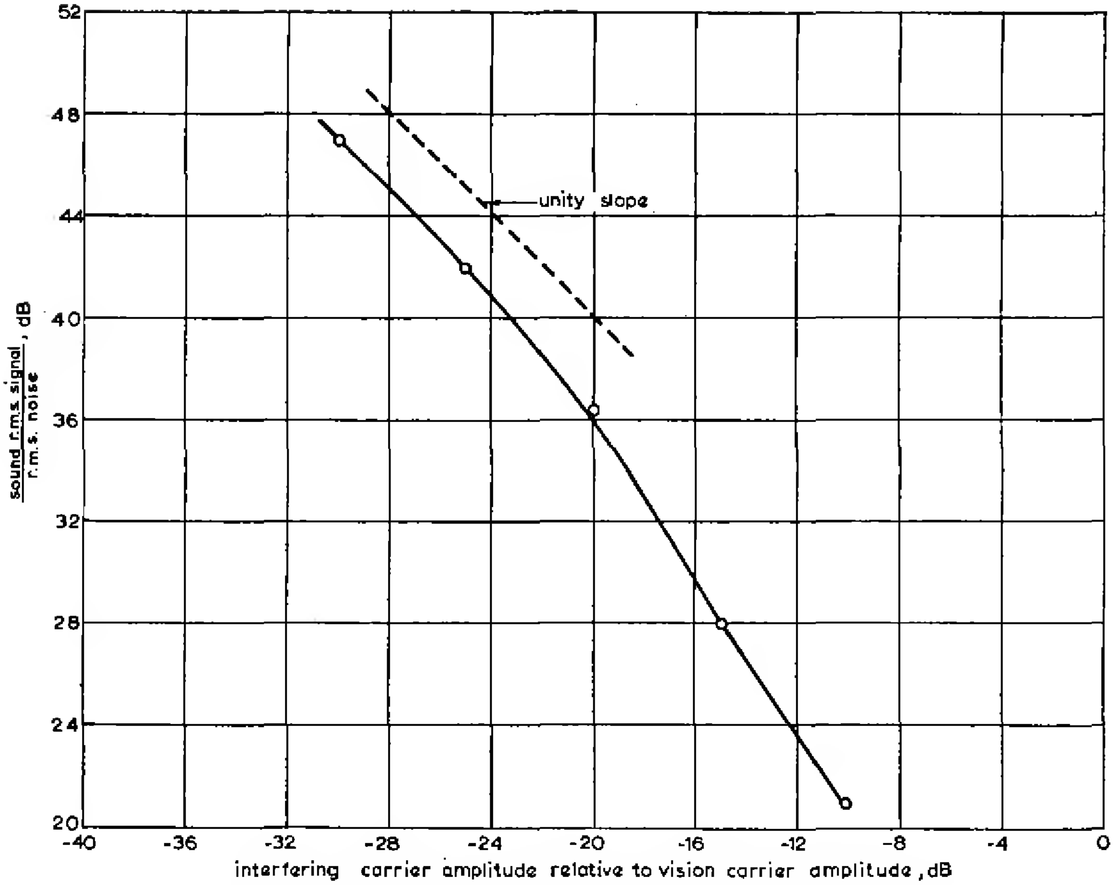

Fig. 15 — The effect of c.w. interference on sound signal-to-noise ratio (40 *per cent deviation*)

3.2.3 *Spurious Sound Signals Caused by the Presence of C.W. Interference*

An unmodulated radio-frequency carrier of adjustable frequency and amplitude was added to the vision-modulated carrier. The two carriers were offset in frequency by 1 kHz which produced a spurious sound signal with a frequency of 1 kHz. This spurious signal is produced by the beat between the wanted and unwanted carriers and it was found that it was principally due to a periodic change in the pulse shape; the change in pulse shape occurs since the interfering signal alters the depth of modulation of the carrier by the sound pulse and, due to the use of vestigial sideband reception, alters the degree of distortion of the pulse. In this way the beat between the two carriers gives rise to a periodic error in the determination of the sound-pulse position. Fig. 15 shows the ratio of the wanted sound output corresponding to 40 per cent of maximum pulse deviation to the spurious sound signal, as a function of interfering-carrier level; the amplitude of the interfering carrier is shown relative to that of the wanted carrier at sync-tip level. It will be seen that the relationship is significantly non-linear at levels of interfering carrier greater than about − 20 dB, the change in sound signal-to-noise ratio being more than the corresponding change in vision signal-to-interference ratio; this is caused by the shape of the sound pulse changing significantly at levels of interfering carrier which are comparable with the low value of vision carrier that occurs at white level (negative modulation).

In the case of an offset in carrier frequencies of two-thirds line-scan frequency, which is used to reduce the visibility of co-channel interference, a 5·2 kHz spurious signal is produced at a similar level to that for a 1 kHz offset.

The optimum offset from a pulse-sound point of view is half the line-scan frequency, when a 7·8 kHz spurious output is produced and heavily attenuated by the decoder output-filter.

3.2.4 *The Effect of the Presence of an Echo*

Echos delayed with respect to the original signal by more than about 2 μs cause an additional impairment to the picture in the form of visible delayed versions of the sound pulses; they appear as a vertical narrow line moving about its mean position in a manner dependent upon the audio signal. A full subjective assessment of this form of picture impairment has not been made but preliminary tests suggest that the visibility of the sound-pulses produces an increase in the picture impairment which is significant but not serious.

When an echo is present, two possible forms of degradation of the sound are possible. The first is due to imperfect

timing of the gating pulses which are used to extract the sound pulses; this may affect the separation of the sound pulses from the video waveform. The second is due to delayed versions of other waveform features which appear on the 'back porch' and cause a discontinuous displacement of the pulse position as it is deviated by modulation. This last effect is shown in Fig. 16 and would be expected to give rise to harmonic distortion or spurious tones.

Fig. 16(*a*) shows a photograph of an echo-free waveform with the sound pulse deviated 60 per cent of maximum by a 1 kHz tone; Fig. 16(*c*) shows the decoded audio output. Fig. 16(*b*) shows an added positive (in-phase) echo with a relative delay of $7 \cdot 0$ μs and a magnitude equal to 20 per cent of the main signal; this choice of delay was such as to place the delayed version of the trailing edge of each line-sync pulse at the undeviated position of the sound pulse. The decoded output is shown in Fig. 16(*d*) and it will be seen that there is no distortion visible. Measurements of distortion were attempted, but only showed that any change in harmonic content was less than $0 \cdot 1$ per cent. In the particular decoder used, echoes of greater than 20 per cent relative magnitude gave rise to loss of reliable sound-pulse gating which produced serious degradation of the sound. This effect does not occur as readily in another type of decoder which uses a self-gating technique; in this type, echoes of the order of 50 per cent may be tolerated. For echoes with relative magnitudes greater than 20 per cent many receiver line-flywheel circuits are affected and picture synchronizing is seriously impaired.

Long-term echoes which cause delayed versions of picture signal to occur in the sound-pulse 'time-slot' give rise to low-level tones depending on picture content. The level of these tones has not been measured but they do not appear to cause significant impairment.

3.2.5 *Performance with Impulsive Interference Present*

The system is susceptible to impulsive interference and any decoder must incorporate suppression. Suitable techniques have been developed and a high degree of suppression can be achieved with comparatively simple circuits while virtually complete suppression is possible with some complication.

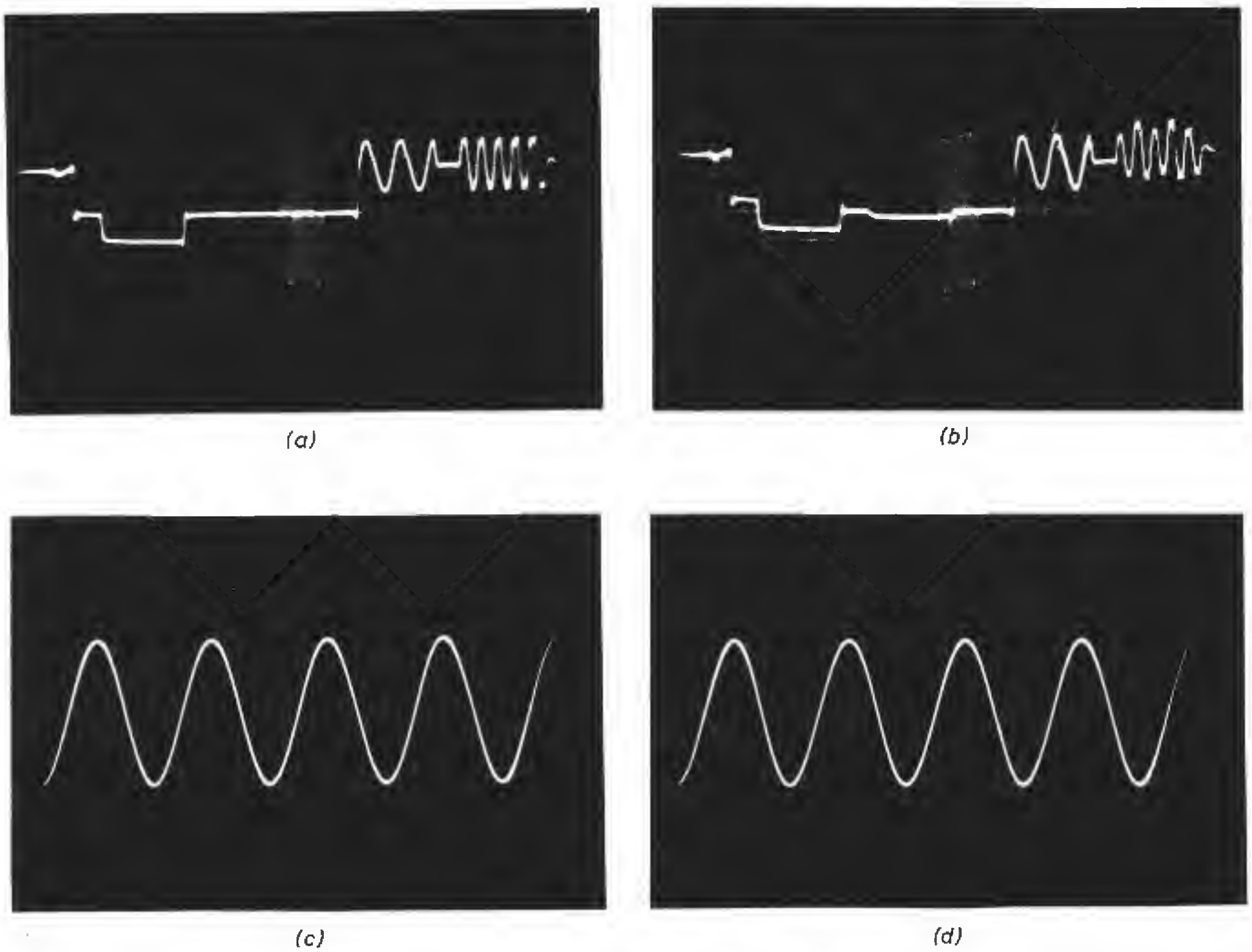

Fig. 16 — The effect of an echo added to the pulse-sound waveform

(*a*) No echo (*b*) 20 per cent positive echo (*c*) Decoded output—no echo (*d*) Decoded output—with echo

4. Conclusions

A pulse-sound system employing one pulse per line-blanking interval has been described which is incapable of broadcasting an audio signal with a bandwidth greater than about 7·0 kHz. This inherent limitation is not crucial and the advantages of the system from the point of view of saving channel bandwidth are considerable.

Compatibility with a representative selection of receivers has been found adequate and could allow the possibility of an early introduction of the system in the u.h.f. bands.

The measured distortion and inversion tones are not serious and could, if necessary, be reduced to a very low level by the use of pre-correction techniques.

In the presence of random noise, the system is not significantly worse in performance than the present a.m. sound accompanying 405-line transmissions in the v.h.f. bands.

C.W. interference causes negligible spurious sound signals unless the vision signal is badly affected.

Echoes cause no significant spurious signals or distortion to the sound until they are of sufficient amplitude to cause failure of line-synchronization in many present-day receivers; the visibility of a sound pulse on the picture, however, constitutes an additional picture degradation.

Impulsive interference suppression must be incorporated in decoders and can be made effective.

From the laboratory tests described it would appear that this system could provide a satisfactory alternative to the present methods of sound transmission with a saving of at least 1 MHz in channel bandwidth.

5. References

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APPENDIX

THE SPECIFICATION OF THE COMPOSITE SIGNAL

The Sound Pulse

The pulse shape is shown in Fig. 7, the total pulse duration being 750 ns. Its energy is substantially contained within the bandwidth 0 to 3·0 MHz and the frequency spectrum is shown in Fig. 8. The peak-to-peak magnitude of the pulse is 1·4 V relative to a 1·0 V composite video signal.

Pulse Position

The centre zero-crossing of the unmodulated pulse is positioned 7·0 μs after the trailing edge of each line-synchronizing pulse.

Pulse Deviation

The audio signal is bandwidth limited to 7·8 kHz and produces deviations of the pulse which are proportional to instantaneous values of the audio signal. These instantaneous values are determined by the 'natural sampling' process in which the instants of sampling the audio signal coincide with the actual timings of the pulses. The deviation is $\pm$ 1·5 μs.

Residual Phase Modulation

Phase modulation of the sound pulse due to timing errors in their generation must be small. A figure of 15 ns peak-to-peak is suggested as an upper limit; this would produce unwanted output at a level of -46 dB with respect to the output produced by 100 per cent audio modulation.

Duration of Line-blanking

The nominal line-blanking interval is extended by 3·5 μs to a total of 15·5 μs by increasing the 'back-porch' duration. 'Front porch', line sync, and colour burst are unchanged in position and duration.

Modification to Field-blanking Waveform

Periods of black level lasting a nominal 9·3 μs are added during the broad pulses, see Fig. 5. These periods, or 'platforms', start at 4·7 μs after the leading edge of each broad pulse.

Audio-signal Pre-emphasis

Audio pre-emphasis, which increases approximately linearly with frequency may be used, so as to increase the deviation at 7·8 kHz by about 3·9 dB. This figure is convenient since a decoder employing a 'sample-and-hold' circuit imposes a 'sin x'/x frequency response upon the recovered audio signal, producing a loss of 3·9 dB at 7·8 kHz.

POSTSCRIPT TO MONOGRAPH No. 67

The introduction to this monograph has recalled that 625-line broadcasting in the U.K. uses channels in u.h.f. which are wider than those used in the v.h.f. bands for 405-line broadcasting. The replacement of the 405-line v.h.f. transmissions by 625-line transmissions using such wide channels would appreciably reduce the number of channels available and as a result it would not be possible to provide substantially full v.h.f. coverage of the U.K. with two 625-line programmes. However, the channel width required for 625-line transmissions could be reduced if the conventional form of sound transmission were replaced by another in which the sound signal may be conveyed as part of a composite vision-and-sound signal.

The object of the work described was to suggest means of effecting a saving in the bandwidth required for radio-frequency channels for the transmission of 625-line television signals in the v.h.f. bands. The adoption of such means would eliminate the need for separate sound transmissions and a saving of bandwidth of about 1 MHz in each channel allocation could be effected.

Introduction of such a system would, however, have meant use of three forms of sound modulation which could have brought difficulties for the receiver industry, while interference effects, particularly sporadic E, would make it difficult to use a channel width in the U.K. different from that in Europe at a time when colour transmissions were in use.

It was therefore decided to abandon work on this proposal as far as broadcasting to the public is concerned, although of course the proposal is of possible interest for other applications.

A RECENT BBC TECHNICAL SUGGESTION

IMPROVING THE LINEARITY OF THE STEEPER SECTION OF A SAWTOOTH WAVEFORM

It is sometimes required (as in phase-discriminator circuits) to generate a sawtooth waveform whose steeper section is substantially linear. Two well-known methods of generating a sawtooth waveform are illustrated in Figs. 1(*a*) and 1(*b*) respectively.

In each case a transistor *TR* is alternately cut off (thereby generating the longer section of the sawtooth) and bottomed (thereby generating the shorter section of the sawtooth). *C* is assumed to reach a maximum voltage V_{max} during each cycle.

In case (*a*) the transistor is assumed to be part of a multivibrator circuit, and discharges *C*, through a predominantly resistive impedance *R*, towards a voltage $V = iR$. In case (*b*) the transistor is assumed to function as a blocking oscillator, and to discharge *C*, through the predominantly inductive impedance, *L*, of the blocking oscillator transformer, towards earth potential. In neither case is the discharge linear; in (*a*) it is exponential, and in (*b*) it is sinusoidal, beginning approximately at the maximum positive value.

The present improvement consists of modifying the discharging impedance (in series with the transistor collector) to a suitable parallel combination of inductance and resistance. Thus, in case (*a*) an inductor is connected in parallel with *R*, and in case (*b*) a resistor is connected in parallel with *L*.

Both circuits then become, during discharge, damped parallel resonant circuits (Fig. 1(*c*)). By suitable choice of the resonant frequency of *L* and *C* (relative to the inverse of the discharge time) and of the resistance *R* (relative to the impedance of *L* or *C* at the resonant frequency), a substantially linear discharge is obtained.)

In case (*a*), without the use of a linearizing inductor, the exponential discharge is towards a voltage $V = iR$, and substantially linear discharge can, of course, be achieved by restricting the magnitude of the sawtooth to a small fraction of $(V_{max} - V)$. Use of a linearizing inductor, however, allows *C* to be discharged through a greater range of voltage for a given non-linearity.

In case (*b*), without the use of a linearizing resistor, the sinusoidal discharge can be arranged to include a substantially linear portion, but always begins with a highly non-linear portion. Use of a linearizing resistor, however, allows substantial linearity to be achieved throughout the discharge.

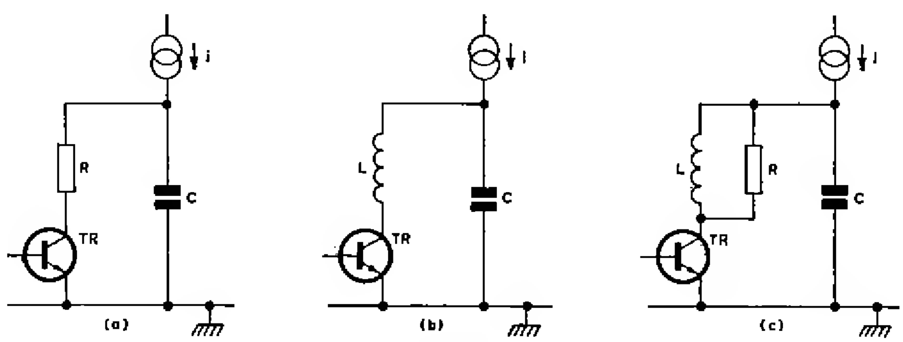

Fig. 1.

W. K. E. Geddes

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The following manuals by members of the Engineering Division of the BBC have been prepared primarily for the Corporation's operating and maintenance staff. They have been made available to a wider public so that the specialized knowledge and experience contained in them may be open to all interested in the engineering side of sound and television broadcasting.

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